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Patentanmeldung Nr. | Patent application No. | Demande de brevet n°

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Der Präsident des Europäischen Patentamts;
Im Auftrag

For the President of the European Patent Office

Le Président de l'Office européen des brevets
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Blatt 2 der Bescheinigung
Sheet 2 of the certificate
Page 2 de l'attestation

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A/D converter with integrated biasing for a microphone

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A/D converter with integrated biasing for a microphone

The invention relates to a combination of a microphone and an A/D (analog-to-digital) converter circuit. Such a combination finds application in, inter alia, hearing aids and mobile telephone sets.

In particular the invention relates to a combination of a microphone requiring a bias signal generated externally, like an electret microphone and an A/D converter circuit. Electret microphones are ideally suited for portable devices, in particular by their fidelity and their small size and weight. To be able to operate electret microphones require the supply of a bias signal generated externally. Usually electret microphones are packed with a JFET in a single housing.

To make use of the generally attractive properties of an electret microphone one has to provide for external generation and supply of a bias current to an electret microphone.

International Patent Application WO-A-96/17493 discloses a combination of a microphone requiring a bias signal generated externally and an A/D converter circuit, of which the input is connected to the output of the microphone for converting the AC signal generated by the microphone to a digital signal at its output.

Although this document does not disclose the circuits needed to provide the bias current to the microphone, it is known to use a separate circuit therefor. Figure1 discloses such a prior art circuit. In this circuit use is made of four external components: two resistors and two capacitors. These are circuit elements, which cannot easily be incorporated into an integrated circuit. Further, it is complicated to accurately control the value of those elements.

The aim of the invention is to provide a combination wherein the disadvantages of the prior art are avoided.

This aim is reached in that the A/D converter is operable to supply a bias signal to the microphone.

This feature eliminates the separate supply and the four extra components required for connecting the power supply to the microphone. Also the need for an extra pin for power supply of the microphone is avoided.

According to a first preferred embodiment the A/D converter is a ΣΔ converter comprising a loop filter, a sampler, a first feed back circuit for ac signals including a first D/A converter and a second feedback circuit for dc signals including a second D/A converter, wherein the bias signal for the microphone is derived from the second feed back circuit, wherein the first and the second feedback circuit are both connected to the input of the integrator circuit .

Obviously, this embodiment entails the well-known advantages of a ΣΔ modulator, like simplicity and low requirements for the accuracy of the components. Usually, the loop filter is implemented as an integrator.

But apart from the advantages, it appears that a ΣΔ modulator is also advantageous for the supply of a bias to a microphone. Indeed, an A/D converter using a ΣΔ modulator comprises a feed back circuit. The feed back circuit is needed for the feedback of the AC output signal of the ΣΔ modulator, but also for feedback of the DC output signal of the modulator. In the prior art the feedback DC signal is used for compensation of any DC offset on the input of the ΣΔ modulator. However, the presence of such a DC feedback loop offers the possibility of supplying an extra power consumer, like a microphone. Of course, the feed back loop should be dimensioned to supply the required DC current to the microphone.

Another advantage of this configuration is that no extra connections between the ΣΔ modulator and the microphone are needed; for the power supply the same two connections can be used as for the AC signal.

Further, it is noted that the microphone delivers a signal as a current. The resistors of the external circuit convert this signal into a voltage. The ΣΔ modulator uses an integrator circuit with a capacitor, which means that the signal has to be converted into a current again in order to be integrated in the capacitor. This is an inefficient way to treat a signal, requiring many components, and leading to the addition of noise generated in these components. The embodiment avoids these disadvantages.

According to a further preferred embodiment, the second feed back circuit comprises a low pass filter with a cut-off frequency, which is lower than the lowest signal frequency of the A/D converter. This feature allows a proper separation between the AC feedback loop and the DC feedback loop, so that both loops can be optimized for their purposes.

This optimization is used in the situation wherein, according to another embodiment, the gain of the second feed back loop comprising the loop filter, the sampler and the second feed back circuit is several orders of magnitude greater than one. In this configuration the aim of the greater loop gain of the DC loop is to provide sufficient supply current to the microphone, so that it can function properly, while maintaining its aim to compensate for off-set at the input of the ΣΔ modulator.

According to another advantageous embodiment the low pass filter is a digital filter and that it is incorporated in the second feed back circuit before the second D/A converter. A digital filter is better suited for implementation in an integrated circuit, as it avoids the problems connected with capacitors, resistors and inductances needed in an analogue filter, when these components have to be integrated in an integrated circuit.

In another preferred embodiment the first and the second feed back circuit are combined to a united feed back circuit incorporating a single D/A (digital-to-analog) converter, of which the input is connected to a low pass filter and a bypass circuit, bypassing the low pass filter. This circuit eliminates the need for a second D/A converter.

In a further preferred embodiment the integrator comprises a bridge circuit, of which the branches comprise current sources, of which circuit a first pair of opposite junctions is connected to a power supply, and of which a second pair of opposite junctions is mutually connected by a capacitor and the microphone, and the junctions of said second pair are each connected to the inputs of the sampler circuit, and wherein a pair of opposite current sources is controlled by the output signal of the feedback circuit. This configuration is very suitable for integration into a single chip, as most elements can be embodied as semiconductor devices.

Subsequently, the present invention will be elucidated with reference to the accompanying drawings, in which:

Figure 1 is a diagram of a prior art circuit for biasing an electret microphone;

Figure 2 is a diagram of a prior art ΣΔ modulator;

Figure 3 is a diagram of a ΣΔ modulator according to a first embodiment of the present invention;

Figure 4 is a diagram of the ΣΔ modulator depicted in Figure 3, but wherein the integrating circuit has been shown in greater detail; and

Figure 5 is a diagram similar to Figure 4, but wherein the D/A converter in the feed back loop has been combined for AC and DC signals.

In applications with a speech input such as mobile telephones and hearing aids often electret capacitor microphones are used. It is common to include a JFET in the housing of such an electret microphone. The gate of the JFET is connected with one of the terminals of the microphone and its drain is connected to the second terminal of the microphone. Further, a bias resistor incorporated in the same housing is connected in parallel to the microphone. The JFET is a depletion device implying that it delivers a DC current if its gate-source voltage $V_{GS} = 0V$. To obtain an output signal representing the variations of the air pressure in its vicinity, the combination of electret microphone and JFET requires a bias current.

Figure 1 shows a diagram of a common prior art circuit to supply such a bias current to the combination of a microphone 1, a bias resistor R_{BIAS} and a JFET (junction field effect transistor) 2, located within a housing 3. This circuit comprises two external resistors R_1 and R_2 to supply the bias current from the power supply V_{CC} and two capacitors C_1 and C_2 to couple the signal to the inputs of a subsequent circuit, like a non-depicted amplifier.

Usually, the DC bias current is about 10 to 50 times larger than the actual AC signal current. The bias resistor R_{BIAS} biases the gate of the JFET, so that its gate-source voltage $V_{GS} = 0$ V. In a typical application the combination of JFET and microphone delivers a current of 300 μA, which is converted into a voltage by the resistors R_1 and R_2. In a substantial number of cases the subsequent circuit comprises an integrator with a capacitor. In that case a voltage signal has to be converted back into a current signal to be able to be integrated on the capacitor of the integrator. Consequently, two opposite conversions are needed in this prior art circuit.

According to the invention a circuit is proposed, in which these conversions are eliminated and wherein the microphone, together with the JFET receives its bias from the subsequent circuit.

To be able to do so said subsequent circuit should comprise a feedback circuit. This feedback circuit is used to supply the bias current. Such a feedback circuit is present in ΣΔ modulators, which are very well suited for processing the signal output of the microphone, in particular for A/D-conversion (analog-to-digital conversion) thereof. It is, however, conceivable that other types of circuits may be used with the present invention.

Figure 2 shows a diagram of a ΣΔ modulator 4. The ΣΔ modulator 4 comprises a filter G1, a sampling device sampling at a rate of m.fs, wherein fs is the Nyquist sampling rate and m is the oversampling factor, and a quantiser circuit 5. Generally, the filter may be digital or analogue, but in the present application the filter is analogue. The output signal of

the quantiser circuit 5 is fed back through a D/A (digital-to-analog) converter 6 and added to the output of the combination of the microphone 1 and the JFET 2 and the resulting added signals are fed to the input of the filter G1. Such a circuit is per se known from the prior art.

Figure 3 shows a diagram of a circuit in which the invention is applied. This circuit comprises a feedback loop which is separated in a feed back loop for AC signals comprising a D/A converter 7 and a feedback loop for DC signals comprising a D/A converter 8. In this respect it is noted that for this purpose also slow varying signals are regarded as DC signals.

To separate the DC signals from the AC signal, a filter G2 is used. This filter G2 is incorporated into the DC feedback loop. As the filter G2 is located before the D/A converter 8, a digital filter has to be used. The cut-off frequency of the DC filter must be lower than the lowest signal frequency. In telephony applications the signal bandwidth is 300-3400 Hz, so that the cut-off frequency should be lower than 300 Hz.

In this circuit the output of the DC feedback loop, that is the output of the D/A converter 8 is applied to the combination of a microphone 1 and JFET 2 as the bias. Of course the bias supplied by the DC feedback loop causes a DC offset in the output signal of the $\Sigma\Delta$ modulator. To minimize this offset the loop gain of the DC feed back loop should be chosen high.

The circuit further comprises the feedback loop for the AC signals. This feedback loop comprises the D/A converter 7. This loop is used for the normal functioning of the $\Sigma\Delta$ modulator.

An implementation example of a first order $\Sigma\Delta$ modulator is shown in Figure 4. It comprises an integrator 9, a sampler/quantizer 5, and separate feedback paths for DC and AC signals. A higher order can be made by increasing the order of the continuous-time loop filter, for example using Gm-C integrators. The AC loop with the AC-D/A converter 7 is a $\Sigma\Delta$ modulator with the subtraction point of input and feedback current at the sources of FETs M_1 and M_2. The feedback current I_{AC} can be positive or negative, depending on the output bitstream code of the modulator.

The DC loop provides feedback only for very low-frequency or DC signals. The bandwidth of the DC loop is determined by filter $G_2(z)$.

The common-mode output voltage of the differential integrator is controlled using common-mode amplifier A_{COM} and controlled current sources I_{COM1} and I_{COM2}. It is assumed here, for simplicity, that this common-mode voltage is correct for $I_{COM1}=I_{COM2}=0$. In

that case, if the microphone current I_{MIC} contains a DC component I_0 and a signal component i_{signal},

$$I_{MIC}=I_0+i_{signal}=I_{DC}-I_1+av(I_{AC}) \quad (1)$$

where $av(I_{AC})$ denotes the average value of switching current source I_{AC}.
The separation in the feedback loop between AC and DC signals forces

$$I_0=I_{DC}-I_1 \quad (2)$$

$$I_{signal}=av(I_{AC}) \quad (3)$$

The output of the ΣΔ modulator is a voltage, and the D/A converters 7 and 8 provide an output current. Suppose the voltage-to-current conversion of the AC-D/A converter 7 has transconductance $g_{m,AC}$, and the DC-D/A converter 8 has transconductance $g_{m,DC}$. The quantizer is, as usual in ΣΔ modulators, modeled by a gain c and an additive noise source N. Furthermore, the loop filter of the ΣΔ modulator, which is of first order in the above example, has a transfer function $G_1(s)$. For simplicity, for low frequencies the effect of the hold function in the D/A converters can be neglected, and the transfer function of $G_2(z)$ together with the hold function is assumed to be the continuous-time transfer function $G_2(s)$. In the frequency domain, the output signal Y of the ΣΔ modulator is:

$$Y=\frac{cG_1}{1+cG_1(g_{m,DC}G_2+g_{m,AC})}I_{MIC}+\frac{1}{1+cG_1(g_{m,DC}G_2+g_{m,AC})}N \quad (4)$$

For frequencies within and below the signal band, so for frequencies much lower than the oversampling frequency mf_s, $cG_1(g_{m,DC}G_2+g_{m,AC}) >> 1$. For signals within the bandwidth of the DC filter also $g_{m,DC}G_2>>g_{m,AC}$, so that

$$Y_{DC}=\frac{1}{g_{m,DC}G_2}I_{MIC}+\frac{1}{cg_{m,DC}G_1G_2}N \quad (5)$$

For frequencies within the signal band of the ΣΔ modulator, so above the cut-off frequency of the DC path, but still much lower than mf_s, $g_{m,DC}G_2>>g_{m,AC}$, so that

$$Y_{AC} = \frac{1}{g_{m,AC}} I_{MIC} + \frac{1}{cg_{m,AC}G_1} N \qquad (6)$$

Combining equations (5) and (6), and assuming that the microphone current I_{MIC} consists of DC component I_o and AC component i_{signal}, then the output signal Y is:

$$Y = \frac{1}{g_{m,DC}G_2} I_0 + \frac{1}{g_{m,AC}} i_{signal} + \frac{1}{cg_{m,AC}G_1} N \qquad (7)$$

The noise term in equation (5) is much smaller than the noise term in equation (6), and is therefore neglected.

Equation (7) shows the offset component in the output signal, which has to be suppressed by sufficient gain in the DC path. The cut-off frequency of the DC path is the frequency for which $g_{m,DC}G_2=g_{m,AC}$.

This technique requires only two IC pins and no external components. Compared to Figure 1, an IC pin and 4 components are saved. Also, this configuration can operate at a very low supply voltage, because the path from the positive supply voltage V_{DD} to the negative supply voltage V_{SS} contains two current sources and a transistor. This requires three times a drain-to-source voltage, which may be as low as 0.4V. Of course, some signal swing is required at the output nodes (the drains of transistors M_1 and M_2 in Figure 4). Also, the gates of transistors M_1 and M_2 must be properly biased. This means that the lower limit of the supply voltage is about 0.7V.

Figure 5 shows another embodiment using only one D/A converter in the feedback loop. This is achieved by adding the AC and the DC feedback signals in the digital domain, just before the DAC. Gain α_{AC} is needed for the appropriate scaling of the AC signal with respect to the DC signal, and is equivalent to $\frac{g_{m,AC}}{g_{m,DC}}$ in the above implementation.

It will be clear that in many ways variations can be applied to the embodiments of the invention discussed above without deviating from the scope of the invention, as expressed in the appending claims.

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CLAIMS:

(54)

1. A combination of a microphone requiring a bias signal and an analog-to-digital converter having an input connected to an output of the microphone for converting a signal generated by the microphone to a digital signal at an output of the analog-to-digital converter, characterized in that the analog-to-digital converter is operable to supply a bias signal to the microphone.

2. The combination as claimed in claim 1, characterized in that the analog-to-digital converter is a sigma-delta converter comprising a loop filter, a sampler, a first feed back circuit for AC signals including a first digital-to-analog converter and a second feedback circuit for DC signals including a second digital-to-analog converter, wherein the bias signal for the microphone is derived from the second feed back circuit, and wherein the first and the second feedback circuit are both coupled to an input of the loop filter.

3. The combination as claimed in claim 2, characterized in that the second feed back circuit comprises a low pass filter with a cut off frequency which is lower than the lowest signal frequency of the analog-to-digital converter.

4. The combination as claimed in claim 2 or 3, characterized in that the gain of the second feed back loop comprising the loop filter, the sampler and the second feed back circuit is several orders of magnitude greater than one.

5. The combination as claimed in claim 3 or 4, characterized in that the low pass filter is a digital filter, and that it is incorporated in the second feed back circuit before the second digital-to-analog converter.

6. The combination as claimed in claim 5, characterized in that the first and the second feed back circuit are combined to a united feed back circuit incorporating a single digital-to-analog converter, of which an input is connected to a low pass filter, and a bypass circuit bypassing the low pass filter.

7. The combination as claimed in claim 2, characterized in that the first integrator comprises a bridge circuit, of which the branches comprise current sources, of which bridge circuit a first pair of opposite junctions is connected to a power supply, and of which a second pair of opposite junctions is mutually connected by a capacitor and the microphone, and the junctions of said second pair are each connected to the inputs of the sampler circuit, and wherein a pair of opposite current sources is controlled by the output signal of the feedback circuit.

8. The combination as claimed in claim 7, characterized in that the integrator comprises a common mode amplifier having an output for driving control inputs of controllable current sources connected between the inputs of the sampler circuit and one of the power supply lines.

ABSTRACT:

A combination of an electret microphone (1) and a sigma-delta A/D converter (9, 5, 7, 8). The sigma-delta A/D converter has a DC feedback loop (8) which provides the bias current (I_{DC}) for the junction FET (2) of the electret microphone. In this way the signal current from the FET (2) can be directly injected in the input integrator (9) of the sigma-delta A/D converter without the need of signal resistors in series with the FET (2).
(Figure 4)

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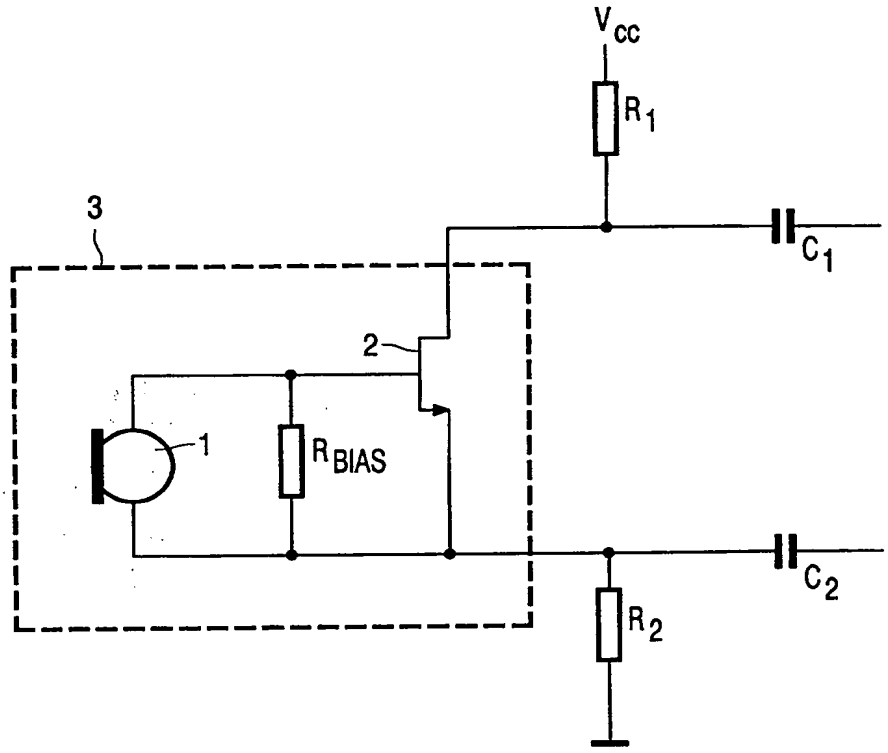

FIG. 1

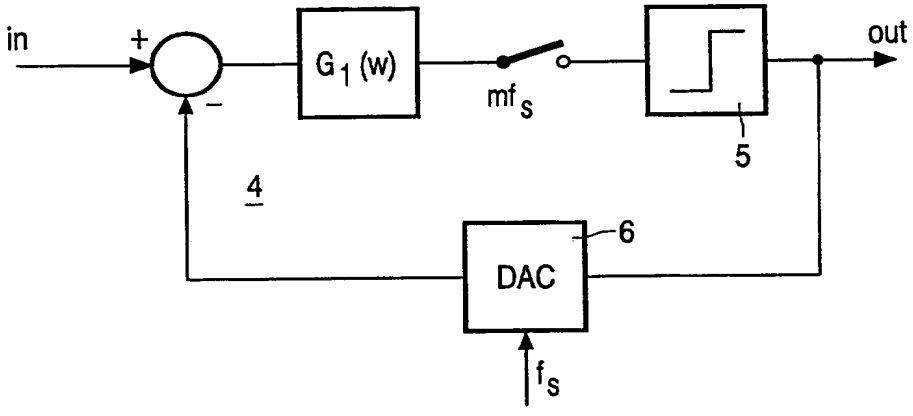

FIG. 2

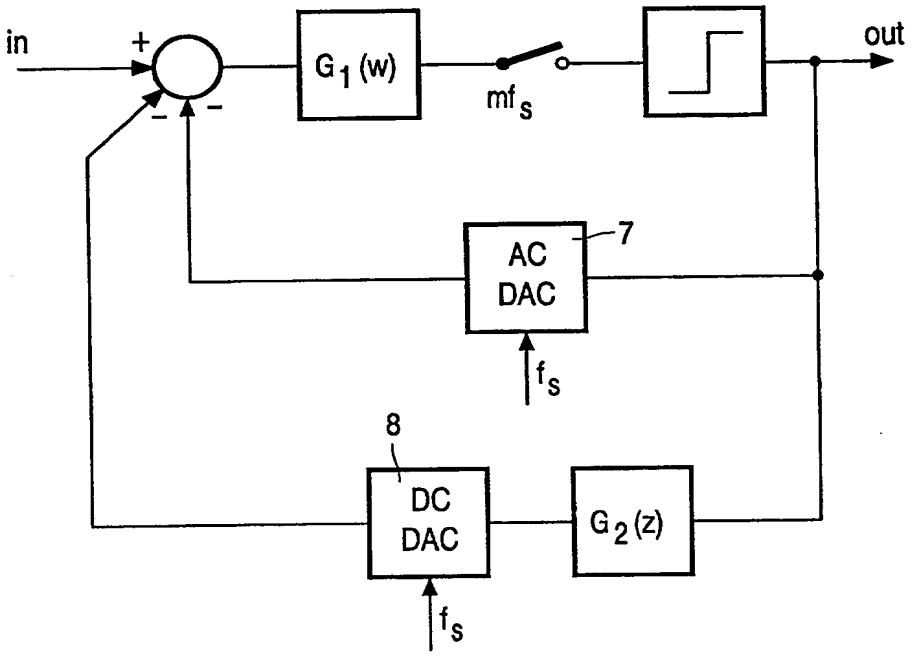

FIG. 3

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FIG. 4

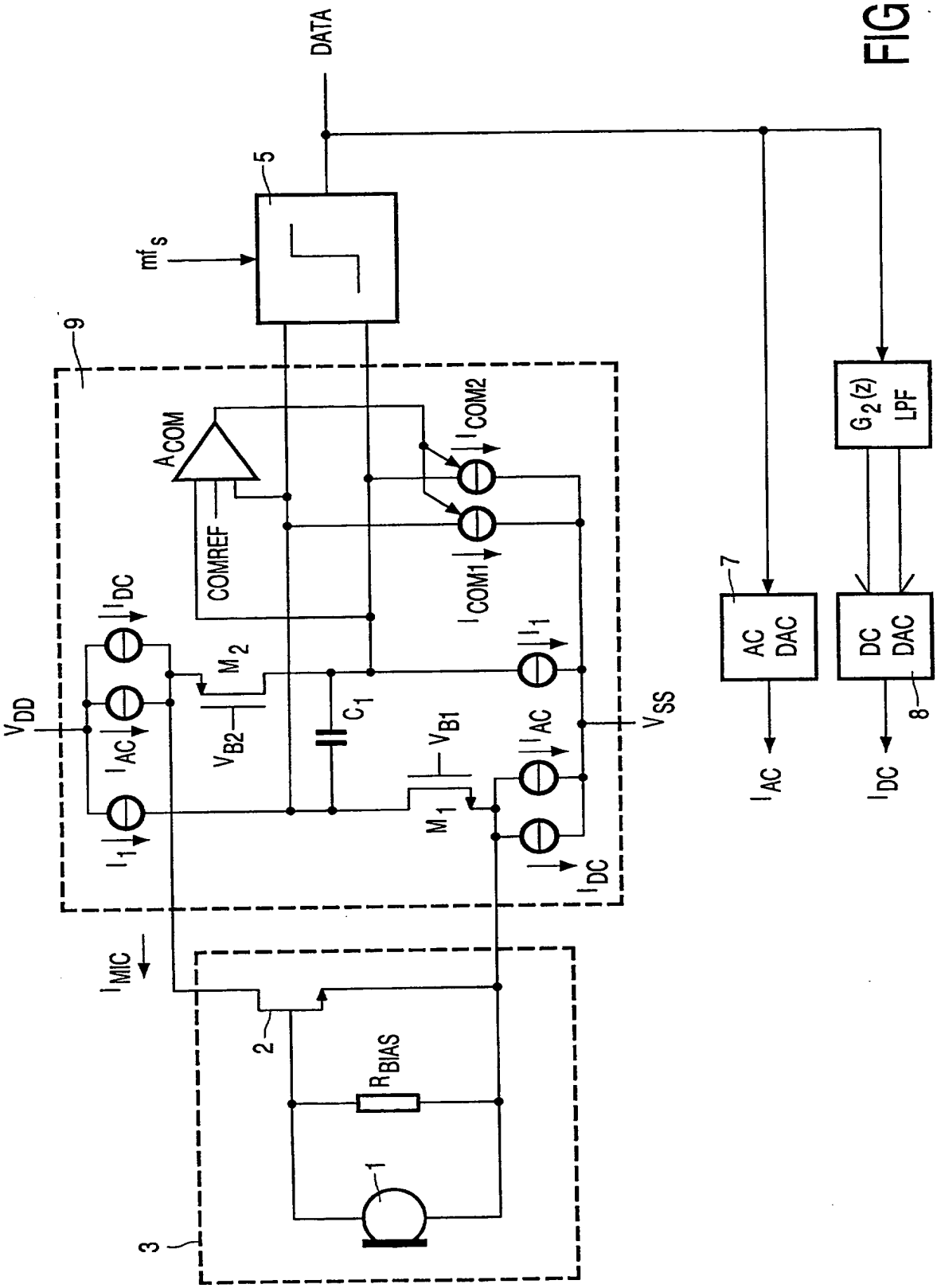
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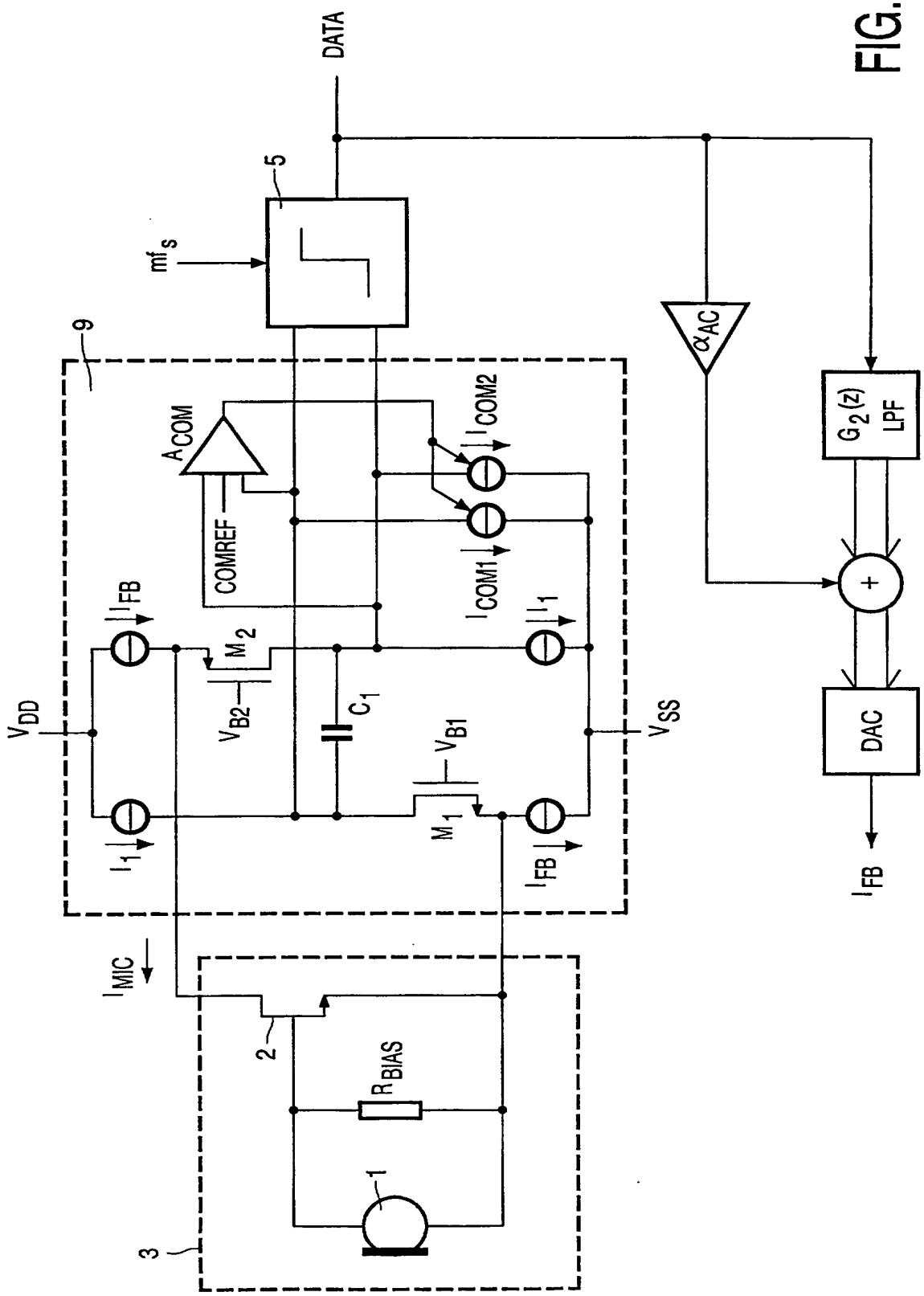
FIG. 5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